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MASSACHUSETTS INSTITUTE OF TECHNOLOGY
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NETWORK SPEECH PROCESSING PROGRAM

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ABSTRACT

This report on the Network Speech Processing Program covers the period 1 October 1977 through 31 March 1978 and reports on the following topics: Voice/Data Integration Study and Demand-Assignment Multiple-Access Techniques.

CONTENTS

NETWORK SPEECH PROCESSING

I. INTRODUCTION

This report documents work at Lincoln Laboratory in the first half of FY 78 on the DCA-sponsored Network Speech program. This program consists of system studies and experiment planning relative to an Experimental Integrated Switched Network. Efforts in two systems studies are reported here. The first is a voice/data integration study directed toward an investigation of the effectiveness of the Slotted Envelope Network (SENET) and SENET Virtual Circuit (SVC) multiplexing techniques. The second relates to the study of Demand Assignment Multiple Access (DAMA) schemes for future integrated satellite networks. In the area of experiment planning, Lincoln's primary effort during this period has been to assist the DCA in specifying the switches, access areas, and satellite communication systems needed for the experimental network. These efforts are not discussed here since the resulting specifications are procurement-sensitive. This report focuses on the system studies mentioned above.

Section II deals with the voice/data integration study and considers the effect of various forms of flow control in a SENET multiplexer. This work builds on a simulation and analysis reported in the previous annual report, in which a basic simulation model was developed and the need for flow control was identified. It is shown here that simple flow control on the data packets in the SENET multiplexer can lead to great reductions in data delay with only small cost in throughput.

Section III represents early efforts on the Demand Assignment Multiple Access study. The focus is on a situation where a large number of ground stations, with a relatively small number of voice users at each station, share a broadcast satellite channel. The problem of designing a DAMA scheme in which full advantage of speech activity detection and not transmitting during silent intervals is achieved via statistical multiplexing at the satellite is set forth, and various alternatives are discussed.

The previous Network Speech Processing Annual Report included work in the Secure Voice Conferencing area. Work in the DCA Secure Voice Conferencing program during FY 78 will be reported under separate cover.

II. VOICE/DATA INTEGRATION STUDY

In the previous Annual Report, the results of a simulation of a Slotted Envelope Network (SENET) voice/data multiplexer were presented. The conclusion was that attempts to achieve high channel utilization led to data-packet queues so large as to overflow any reasonable amount of storage in the multiplexer. Even assuming infinite storage, the mean data delays were large because of the buffer buildup during periods when voice channel occupancy was high. The need for a flow-control mechanism became apparent, and two types of flow control have been investigated: (1) voice-rate control, where a new voice user is assigned (at dial-up) one of several available bit rates, based on the current voice utilization and/or data-queue size; and (2) data-flow control, where a fixed limit is imposed on the size of the data queue.

A set of simulations has been run to gather statistics on multiplexer performance with different flow-control strategies. In all cases, total channel capacity was fixed at 120 kbps, with voice capacity limited to 80 kbps and 40 kbps dedicated to data. A movable boundary was employed to allow data packets to be sent in temporarily unused portions of the voice capacity. The voice calls were assumed to arrive according to a Poisson process at a rate 0.05 call/sec, with exponentially distributed holding times of mean duration = 100 sec, corresponding to 5 erlangs of voice traffic. The maximum number of simultaneous voice calls was limited to 10, with blocked calls cleared. Data packets, with fixed lengths of 80 bits, were assumed to arrive in a Poisson process with variable arrival rate. A packet arrival rate of 500 per second would completely saturate the 40 kbps dedicated to data. The purpose of the simulations was to investigate system performance for data packet arrival rates greater than 500 per second, when a portion of the voice capacity must be utilized for data. The performance measures of interest for data include average delay, average queue size, maximum queue size, and fraction of packets arriving when the data queue is full (when data-flow control is employed). With voice-rate control, the distribution of callers among the available bit rates and the average voice bit rate assigned serve as performance measures for voice users.

A. EXPERIMENTS WITH VOICE-RATE CONTROL BUT NO DATA-FLOW CONTROL

In the previously reported SENET simulation, all voice users were assumed to operate at 8 kbps. The voice channel utilization exhibited large variation around its mean, and data queues would build up during the peaks of voice utilization. The idea behind the voice-rate control techniques studied here was to cut down the peaks of voice channel utilization by assigning lower bit rates to callers who enter the system when utilization is high. The results for four voice-rate control schemes are reported here. The first scheme that was tried is illustrated in Fig. II-1. New calls were assigned at 2, 4, 8, or 16 kbps as a function of current total voice channel utilization, with lower rates assigned during higher utilization periods. The inclusion of a 16-kbps rate would provide better voice service for some users, but also had the effect of filling in the valleys of voice utilization that were present when voice-rate control was employed. The second scheme was similar to the first except that the voice rate of 16 kbps was not used, but 8 kbps was assigned to callers arriving when utilization was less than 20 kbps. The third approach used voice-rate control as depicted in Fig. II-1, except that monitoring of the data queue was included. When the data queue was empty, the assignments indicated by Fig. II-1 were employed, but voice callers arriving when the data queue was not empty were assigned a

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rate of 2 kbps. In the fourth technique, the rate of a new call was governed solely by the size of the data queue. If the queue was empty, a rate of 8 kbps was assigned. If the queue was non-empty but did not exceed 150 packets, 4 kbps was assigned; otherwise, 2 kbps was used. The four approaches just outlined will be referred to below as schemes v1, v2, v3, and v4; v0 will denote the situation where no control is imposed.

Results on average packet delay as a function of data packet arrival rate for the four schemes just described, and for a situation with no flow control, are summarized in Table II-1. Table II-2 shows the average bit rate assigned to a voice user for each case. The first scheme represents an improvement over no flow control for packet arrival rates of 600, 700, and 800. The price for this improvement was a drop in average voice bit rate from 8 to 5.7 kbps. However, the flow control worsened the situation when packet arrivals reached 900 per second. When 16 kbps was eliminated as a voice rate (scheme v2), a general improvement in delay performance was realized. In addition, the average voice bit rate increased to 6.6 kbps because fewer users had to be assigned to the lower rates. Scheme v3 (the same as v1 except that some data-queue monitoring was introduced) represented a significant improvement over v1 in terms of data delay. Scheme v4, where voice-rate control was based strictly on data queue size, was generally the most effective of the voice-rate control techniques.

However, none of the performance results shown in Table II-1 are really satisfactory at the high packet arrival rates. Even for scheme v4, the delay of 0.63 sec at 900 packets/sec is unacceptable, and corresponds to an average queue size of 571 packets. Observed maximum queue sizes were significantly higher. The conclusion is that, although voice-rate control alone can enhance data performance, some form of data-flow control is necessary to keep delays and queue sizes within reasonable limits.

B. EXPERIMENTS WITH DATA-FLOW CONTROL

The first experiment with data-flow control involved limiting the data buffer to a fixed maximum size of 150 packets, but not including any voice-rate control. Data packets are denied entry to the multiplexer when its queue is full. This represents added delay since these packets have to be retransmitted to the multiplexer. If the packets enter the multiplexer directly from a user terminal, then the terminal would retransmit when no acknowledgement was received. Otherwise, a store-and-forward node feeding the multiplexer could handle the retransmission. The second experiment combined the same data-flow control procedure with the data-queue-independent voice-rate control employed in scheme v2 above. The third test combined limitation of the data buffer to 150 packets with data-queue-dependent voice-rate control similar to the method used in scheme v4 above. If the queue was empty, a rate of 8 kbps was assigned. If the queue was non-empty but did not exceed 75 packets, 4 kbps was assigned; otherwise, 2 kbps was used. The three data-flow control techniques just described will be referred to as d1, d2 and d3.

The average packet delays for the three cases are plotted in Fig. II-2. Tables II-3 and -4 show the average voice bit rates and the percentage of data packets discarded for each case. In all cases, the improvement in data packet delay is dramatic as compared to the situation where no data-flow control is imposed. For example, the worst case in Fig. II-2 is an average delay of 49 msec for scheme d1 at 900 packets/sec arrival rate. The corresponding best result without data-flow control is an average delay of 630 msec for v4. In addition, Table II-3 shows that the percentage of arriving packets which find a full data queue is rather low in all cases.

TABLE II-1

AVERAGE DATA PACKET DELAY AS A FUNCTION OF DATA PACKET ARRIVAL RATE FOR VOICE-RATE CONTROL SCHEMES
(v0 – v4 as described in text)

Packet Arrival Rate (packets/sec) \ Flow Control Scheme	vφ	v1	v2	v3	v4
	Average Delay (sec)				
600	0.18	0.0	0.0	0.0	0.0
700	1.18	0.0	0.0	0.0	.12
800	3.08	1.19	.14	.33	.30
900	11.6	39.0	1.6	2.55	.63

TABLE II-2

AVERAGE BIT RATE ASSIGNED TO VOICE USERS FOR VOICE-RATE CONTROL SCHEMES v0 – v4

Packet Arrival Rate (packets/sec) \ Flow Control Scheme	vφ	v1	v2	v3	v4
	Average Voice Bit Rate (kbps)				
600	8.0	5.7	6.6	6.0	7.9
700	8.0	5.7	6.6	6.0	7.6
800	8.0	5.7	6.6	5.5	7.2
900	8.0	5.7	6.6	4.8	6.5

TABLE II-3

AVERAGE BIT RATE ASSIGNED TO VOICE USERS FOR DATA-FLOW CONTROL SCHEMES d1 – d3

Packet Arrival Rate (packets/sec) \ Flow Control Scheme	d1	d2	d3
	Average Voice Bit Rate (kbps)		
600	8.0	6.6	7.9
700	8.0	6.6	7.6
800	8.0	6.6	7.2
900	8.0	6.6	6.8

TABLE II-4

PERCENTAGES OF PACKETS ARRIVING WHEN QUEUE IS FULL UNDER DATA-FLOW CONTROL SCHEMES d1 – d3

Packet Arrival Rate (packets/sec) \ Flow Control Scheme	d1	d2	d3
	Packets Arriving When Data Queue is full (percent)		
600	0.3	0.0	0.1
700	1.0	0.0	0.4
800	2.6	0.2	0.7
900	4.5	1.2	1.1

This implies that the additional delay due to retransmission of these packets also ought to be modest.

If schemes d1-d3 are compared on the basis of data packet delay alone, it is clear from Fig. II-2 that the best performance is achieved with d2, which includes voice-rate control independent of the data queue. However, the improvement in delay performance over d3 is counterbalanced by the fact that higher average voice bit rates were assigned in scheme d3 at all data packet arrival rates. In general, acceptable performance could be achieved with any of the three data-flow control approaches.

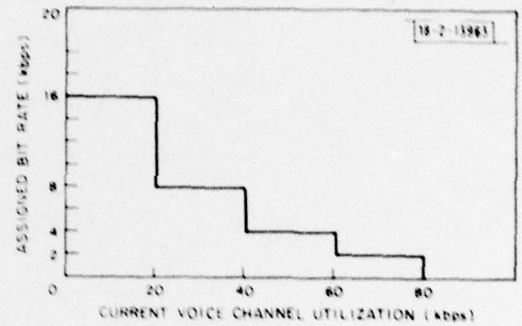

Fig. II-1. Example of voice-rate control based on current voice channel utilization. The bit rate assigned to a new user is plotted as a function of the current total bit rate of all voice users.

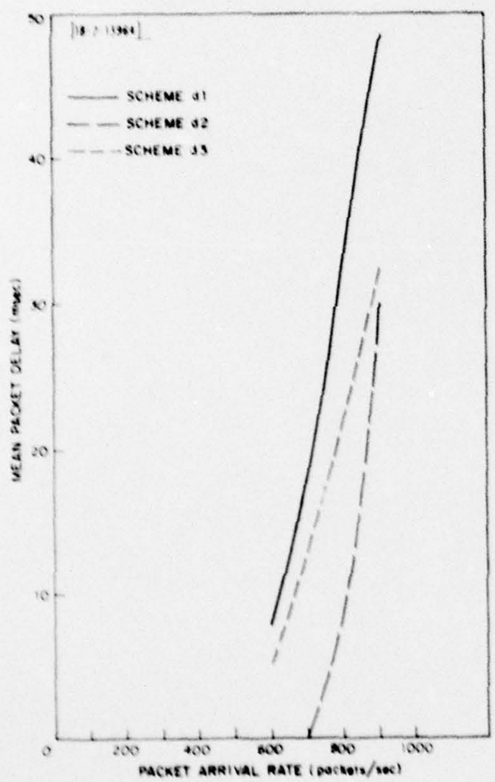

Fig. II-2. Average data packet delay as a function of data packet arrival rate for different data-flow control experiments d1-d3 described in the text.

III. DEMAND ASSIGNMENT MULTIPLE ACCESS TECHNIQUES

One of Lincoln Laboratory's tasks under the Experimental Integrated Switched Network program is to investigate various satellite DAMA schemes to assess their suitability for use with combined voice/data traffic. A problem of particular interest for possible future satellite networks is that of efficiently multiplexing the traffic from a large number of relatively narrowband ground terminals onto a wideband satellite channel. Since the advantage of statistically multiplexing a small number of users is severely limited, the multiplexing will have to be effected at the satellite where the statistical properties of the large, aggregate traffic stream are available.

Figure III-1 depicts a typical satellite voice communication system involving a large number (m) of earth terminals each servicing a small number (5 to 10) of voice communicators. There are several sources of statistical fluctuation inherent in this situation:

(a) The number of active speakers is a random variable.

(b) The length of time a given speaker is actually talking (talkspurt duration) is random.

These statistical effects must be exploited in order to design a system that makes efficient use of the available satellite channel capacity. If the statistical effects were not exploited, the number of voice channels required would be equal to $\sum_{i=1}^{m} = M_i$ where M_i denotes the maximum number of voice circuits available at the i^{th} terminal and m is the number of ground terminals. The speaker talkspurt statistics mentioned above offer a potential reduction of the required number of voice circuits of about 40 to 50 percent. This comes about because speakers, on the average, only speak about 50 to 60 percent of the time. Note that this potential saving cannot be realized at the terminals themselves because, for a small number of speakers, say less than ten, the statistical fluctuations of the talkspurt intervals above their average value are large, thus requiring the number of circuits to be well above the average. The talkspurt advantage can be potentially realized at the satellite, however, because there the number of speakers is large and the statistical fluctuations of the talkspurt intervals about their average value are correspondingly small. Stated somewhat more concretely, the exploitation of talkspurt statistics may allow 100 speakers to use only 60 circuits, but 10 speakers may require as many as 8 or 9 circuits.

In similar manner, the number of required voice circuits can be further reduced by exploiting the random nature of the number of calls actually in progress at a given time. Here the trade-off is between the number of circuits required to serve a given speaker population and the percentage of calls blocked. As an example of the magnitude of the saving that can be obtained by this means, telephone experience shows that if the average number of speakers seeking to talk is 65, then 80 circuits are required to maintain a blocking probability of 0.01.

The most straightforward solution to the above problem is Time Division Multiple Access (TDMA) in which each ground station is assigned a time slot during which it has exclusive use of the satellite channel. This scheme suffers from two major drawbacks:

(a) It does not take advantage of talkspurt statistics, which means that the channel utilization for voice traffic is at best about 60 percent.

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(b) Because of the time-division strategy, each station must transmit at the full satellite bandwidth even though it produces traffic at a much lower bandwidth. This might require an expensive satellite terminal, which would be uneconomical for just a few users.

The use of Frequency Division Multiple Access (FDMA) removes the second of the above objections, but does nothing to allay the first. In fact, the channel utilization will be even less because FDMA usually requires larger guard bands than TDMA.

A variety of schemes for achieving higher channel utilization than simple TDMA have been proposed. Among these are CPODA, Directionally Variable Multiple Access, Fully Variable Demand Access, Binder's Reservation Algorithm, and Crowther's Reservation Aloha. These techniques have been extensively analyzed and/or simulated with regard to their ability to handle data traffic, but little work has been done to assess their suitability for voice. Work is now in progress to remedy this deficiency. Candidate DAMA schemes will be simulated on a PDP-11/45 computer. Queue length statistics, packet loss probability, packet delay statistics, and channel utilization will then be evaluated using measured speaker talkspurt statistics to generate the incoming voice traffic.

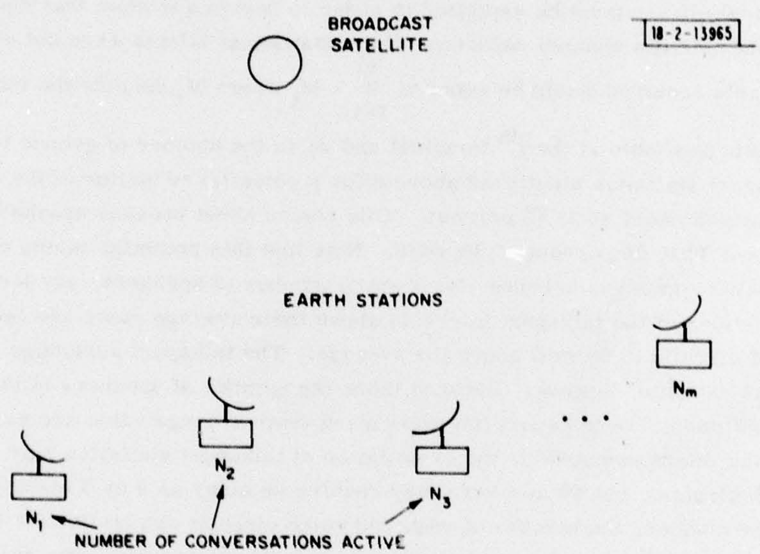

Fig. III-1. Configuration of satellite communication system.

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